WEST Search History

DATE: Friday, September 05, 2003

Set Name side by side	Query	Hit Count	Set Name result set
	DB=USPT,JPAB,EPAB,DWPI,TDBD; PLUR=YES; OP=OR		
L5	(((wide adj band$) or wideband$) near3 (narrowband$ or (narrow adj band$))) and (cascad$3 or serial) near2 (filter$ or equaliz$3 or decimat$3)) and ((band near2 reject$3)	13	L5
L4	((dual$ or multi$4) near2 mode$) and (cascad$3 or serial) near2 (filter$ or equaliz$3 or decimat$3)) and ((band near2 reject$3)	20	L4
L3	((dual$ or multi$4) near2 mode$) and (cascad$3 or serial) and ((band near2 reject$3) near3 (filter$ or equaliz$3 or decimat$3))	32	L3
L2	((dual$ or multi$4) near2 mode$) and (mobile or cellular$) and ((band near2 reject$3) near3 (filter$ or equaliz$3 or decimat$3))	35	L2
L1	((dual$ or multi$4) near2 mode$) same (mobile or cellular$) same ((band near2 reject$3) near3 (filter$ or equaliz$3 or decimat$3))	0	L1

END OF SEARCH HISTORY

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L4: Entry 10 of 20 File: USPT May 26, 1998

DOCUMENT-IDENTIFIER: US 5757858 A
TITLE: Dual-mode digital FM communication system

Abstract Text (1):
A dual-mode digital communication system for communicating an information signal during operation in frequency-modulated (FM) and multiple-access modes is disclosed herein. The digital communication system includes a dual-mode transmitter for transmitting the information signal using an FM communication signal during FM mode operation, and for transmitting the information signal using a multiple-access communication signal during multiple-access mode operation. The communication system further includes a dual-mode receiver for receiving the FM communication signal during FM mode operation, and for receiving the multiple-access communication signal during multiple-access mode operation. Incorporated within the dual-mode receiver is a digital demodulator for recovering the information signal from the received FM signal during operation in the FM mode, and for recovering the information signal from the received multiple-access signal during multiple-access mode operation. In a preferred implementation the dual-mode transmitter is disposed to convert a first sequence of binary data within the information signal into a sampled modulation waveform, and to provide the FM communication signal by modulating a carrier signal based at least in part on the sampled modulation waveform. The dual-mode transmitter may also be configured to multiplex a second sequence of wideband message data with the sampled modulation waveform so as to form a composite FM modulation waveform.

Brief Summary Text (3):
The present invention relates generally to frequency modulated (FM) communication systems. More particularly, the present invention relates to a novel dual-mode communication system selectively operative in either FM or code division multiple access (CDMA) modes.

Brief Summary Text (9):
In certain operating environments, a digital receiver may receive a signal which experiences rapid and wide variations in signal power. For example, in digital receivers such as are used in a code division multiple access (CDMA) mobile cellular telephones, it is necessary to limit the power of the received signal for proper processing of the received signal. Similarly, in digital receivers which are CDMA compatible and conventional FM compatible, i.e., dual-mode CDMA/FM receivers, it is necessary to provide power limiting of both wideband CDMA signals and narrowband FM signals. The limiting process is complicated by the differing dynamic ranges associated with the received FM and CDMA signal power. That is, the magnitude of received FM signals may vary over a dynamic range as large as 110 dB in cellular systems, whereas existing power control techniques within CDMA systems typically result in a much more limited dynamic range, i.e., approximately 80 dB.

Brief Summary Text (10):
The provision of separate AGC circuitry for each of the modes of dual-mode CDMA/FM receivers increases the hardware complexity and expense of such receivers. Accordingly, it would be desirable to provide AGC circuitry capable of operating both upon narrowband, wide-dynamic range FM signals, as well as upon wideband CDMA signals of more limited dynamic range.

Brief Summary Text (15):
Signal filtering is often performed using intermediate frequency (IF) filters in order to achieve adequate out of band signal rejection. Although the requisite signal rejection capability may be achieved through the use of ceramic IF filters, these tend to be relatively large and expensive. Smaller and less expensive IF

filters are generally incapable of being realized so as to possess the desired signal rejection characteristics, and hence are generally not employed in FM cellular telephone receivers.

Brief Summary Text (16):
As is well known, recent advances in integrated circuit (IC) technology have made possible the realization of active baseband filters which are quite small and inexpensive. It follows that it would be desirable to employ active IC baseband filters to effect significant out of band signal rejection, thereby allowing small and inexpensive IF filters to be used to provide any additional required signal rejection. It is also known that the rejection capability of an active filter is a function of filter gain, but that higher gain active baseband IC filters exhibit an increased susceptibility to undesirable DC bias offsets. This increased susceptibility may be attributed to increased amplification of offset sources. That is, these spurious DC signal components are amplified by the active baseband IC filter, and will act as a noise source in the FM demodulator.

Brief Summary Text (19):
It is therefore a further object of the present invention to provide an AGC apparatus, for incorporation within a dual mode receiver, which is capable of effecting baseband filtering without loss of carrier frequency information.

Brief Summary Text (21):
The present invention is a novel dual-mode digital communication system for communicating an information signal during operation in frequency-modulated (FM) mode and code division multiple-access (CDMA) mode. The digital communication system includes a dual-mode transmitter for transmitting the information signal using an FM communication signal during FM mode operation, and for transmitting the information signal using a spread spectrum QPSK signal during CDMA mode operation.

Brief Summary Text (22):
The communication system further includes a dual-mode receiver for receiving the FM communication signal during FM mode operation, and for receiving the spread spectrum QPSK signal during CDMA mode operation. The dual mode receiver further includes a digital demodulator for recovering the information signal from the received FM signal during operation in the FM mode and for recovering the information signal from the received QPSK signal during CDMA mode operation.

Brief Summary Text (23):
With regard to the dual-mode receiver, an FM demodulator may be included within the digital demodulator in order to convert the digitized received baseband FM communication signal, preferably centered about a predefined baseband frequency offset from the zero frequency, into the recovered information signal. The conversion is performed such that each digital sample of the received baseband signal includes first and second sample components in phase quadrature. In a preferred implementation the FM demodulator initially calculates a ratio of the first and second sample components of each digital sample of the input signal. A phase demodulated signal is computed by determining the arctangent of each digital sample ratio. A frequency demodulated signal, equivalent to the demodulated information signal, is computed by filtering the demodulated phase sequence using a digital differentiator.

Drawing Description Text (2):
FIG. 1 provides an illustrative representation of a dual-mode digital communication system in accordance with the invention.

Drawing Description Text (3):
FIG. 2 shows a block diagram representation of an audio interpolation filter included within a dual-mode transmitter of the inventive communication system.

Drawing Description Text (4):
FIGS. 3A-3D illustratively represent a set of sampled, sinusoidal-like wideband waveforms produced by a wideband waveform generator included within the dual-mode transmitter in FM mode.

Drawing Description Text (7):
FIG. 5 shows a block diagram of a combiner & gain adjust network included within the dual-mode transmitter.

Drawing Description Text (9):
FIG. 7 shows a block diagram of a preferred implementation of an I/Q FM demodulator included within a dual-mode receiver of the invention.

Drawing Description Text (19):
FIG. 14A illustratively represents the architecture of a preferred implementation of a frequency tracking loop included within the dual-mode receiver of the invention.

Drawing Description Text (21):
FIG. 15 provides a block diagram of a preferred implementation of an audio decimation filter included within the dual-mode receiver of the invention.

Detailed Description Text (2):
FIG. 1 provides an illustrative representation of a dual-mode digital communication system in accordance with the invention. The dual-mode communication system includes a dual-mode transmitter 14 disposed to transmit digital information signals to a dual-mode digital receiver 16. In a digital FM mode, the transmitted digital information signals are generated within the dual-mode transmitter through FM modulation of an RF carrier on the basis of digitized audio and wideband data signals. During a multiple-access mode of operation, the transmitted information signals comprise, for example, code-division multiple-access (CDMA) information signals. An FM/CDMA mode select signal provided to transmitter 14 by a control processor (not shown) allows selection of either the digital FM or CDMA mode of operation.

Detailed Description Text (3):
I. Overview of Dual-Mode Transmitter

Detailed Description Text (6):
Upon switching from operation in the digital FM mode to operation in the CDMA mode, a CDMA data signal carried by CDMA input line 66 is supplied to the RF transmitter 64 in lieu of the digital FM mode modulation signal. During CDMA mode operation the transmitter 64 generates in-phase (I) and quadrature phase (Q) pseudorandom noise sequences PNI and PNQ, which typically correspond to a particular area (i.e., cell) to which information is transmitted during CDMA mode operation. Within transmitter 64 the CDMA data signal is XOR'ed with the PNI and PNQ sequences so as to spread the CDMA data signal prior to transmission. The resulting I-channel code spread sequence and Q-channel code spread sequences are used to bi-phase modulate a quadrature pair of sinusoids. The modulated sinusoids are summed, band-pass filtered, shifted to an RF frequency, and again filtered and amplified prior to being radiated via antenna 70 over a communication channel to dual-mode receiver 16. An exemplary CDMA transmitter and waveform generator is described in, for example, U.S. Pat. No. 5,103,459, issued Apr. 7, 1992, entitled System and Method for Generating Signal Waveforms in a CDMA Cellular Telephone System, which is assigned to the assignee of the present invention and which is herein incorporated by reference.

Detailed Description Text (7):
II. Overview of Dual-Mode Receiver

Detailed Description Text (8):
Again referring to FIG. 1, the dual-mode receiver 16 includes a receive antenna 80 for receiving the I and Q channel information signals transmitted by dual-mode transmitter 14. During both CDMA and digital FM mode operation, the I and Q channel information signals received by antenna 80 are processed by a direct conversion analog receiver 84. Within the analog receiver 84, the I and Q channel information signals are mixed with a local oscillator signal to produce in-phase (I) and quadrature-phase (Q) baseband signals. During digital FM mode operation the local oscillator frequency is selected to be offset by a predetermined margin from the RF carrier frequency. In this way the received I and Q information signals are converted to I and Q digital FM baseband signals offset from zero frequency, i.e., from a "zero-IF", by the predetermined margin.

Detailed Description Text (13):
III. Detailed Description of Dual-Mode Transmitter

Detailed Description Text (26):
IV. Detailed Description of Dual-Mode Receiver

Detailed Description Text (43):
In particular, the table of FIG. 9B is indicative of the manner in which the bit shifts carried out within shift register 278 set the time constant of the DC offset correction loop. As is indicated by FIG. 9B, the DC offset correction loop is capable of operation in both TRACKING and ACQUISITION modes. Operation in the ACQUISITION mode is characterized by shorter loop time constants, which allows for rapid initial convergence of the offset correction. The relatively short loop time constants utilized in ACQUISITION mode increase the loop bandwidth relative to the TRACKING mode bandwidth, allowing larger offset errors to exist during ACQUISITION mode than in TRACKING mode. Conversely, the longer loop time constants, and hence narrower loop bandwidth, are used during TRACKING mode in order to minimize steady state offset errors. Such dual-mode operation allows initial convergence to be obtained with minimal delay, yet simultaneously enables optimization of steady-state performance.

Detailed Description Text (64):
FIG. 14A illustratively represents the architecture of a preferred implementation of a frequency tracking loop included within the dual-mode receiver. Referring to FIG. 14A, the frequency tracking loop filter 250 includes a digital subtractor 390, to which is provided the 8-bit FM demodulated frequency signal from the phase to frequency generator 218. Subtractor 390 is designed to subtract the frequency bias signal (F.sub.BIAS), which in an exemplary implementation is approximately equivalent to one frequency LSB (i.e., to 156 Hz), from the 8-bit demodulated frequency signal. The resulting difference signal is supplied to a shift register 392, and is bit-shifted in accordance with a frequency tracking loop gain constant F.sub.GAIN. Compiled within the table of FIG. 14B are the time constants of the frequency tracking loop associated with various gain constants F.sub.GAIN.

Detailed Description Text (69):
The filter architecture of FIG. 15 relies on a set of cascaded SINC filters (i.e. sinc(x)=.sup.sin(x) 1/.sub.x) to achieve hardware implementation efficiency. In particular, the filter 102 includes an input 3-tap SINC filter 412 cascaded with a 2-tap SINC.sup.3 filter 416. The output from the SINC.sup.3 filter 416 is sub-sampled by switch 418 at the exemplary rate of 20 ksps. The filter 102 will typically be designed to provide at least 40 dB of attenuation over the frequency range from 16 to 20 kHz. This degree of attenuation may be effected by realizing the SINC filters in accordance with the following z-domain transfer functions:

Detailed Description Text (71):
Turning now to FIG. 16, there is illustratively represented the architecture of an exemplary implementation of the wideband data recovery network 104. The wideband data recovery network 104 is disposed to perform timing recovery and decoding operations upon the Manchester-encoded data stream received by the dual-mode receiver 16. The network 104 includes a receiver (RX) filter 430, which approximates a matched filter for the Manchester symbol stream. In addition, the RX filter 430 band-limits the demodulated FM signal and rejects any high-frequency noise produced by the phase to frequency generator 218. In an exemplary implementation the RX filter 430 is designed to emulate the characteristics of an analog 4.sup.th -order Butterworth low-pass filter having a cut-off frequency of approximately 13 kHz. Given that the demodulated FM signal is generated by the I/Q demodulator at an exemplary 40 ksps rate, the RX filter 430 may be realized as a 2-tap SINC filter (zero at 20 ksps), having a z-domain transfer function of (1+z-1)/2.

Detailed Description Text (86):
In a preferred implementation the serial stream of NRZ data from output register 566 may include messages of various formats encoded therein by the dual-mode transmitter 14. It is anticipated that techniques may be devised by those skilled in the art for identifying and extracting such message information from the serial NRZ data.

CLAIMS:

1. A dual-mode transmitter for transmitting an information signal using a frequency modulation (FM) signal when in an FM mode, and using a code division multiple access (CDMA) signal when in a CDMA mode, comprising:

a digital signal processor for generating a digital FM audio signal;

a wideband data generator, coupled to said digital signal processor, for generating a wideband data signal said wideband data generator comprising

a data register for receiving a non-return to zero input data signal;

a first multiplexer, coupled to said data register, for generating a portion of said wideband data signal in response to said non-return to zero input data signal;

an inverter, coupled to said first multiplexer, for inverting said generated portion of said wideband data signal; and

a second multiplexer, coupled to said inverter, said data register and said first multiplexer, for multiplexing said portion of said wideband data signal with said inverted portion of said wideband data signal, thereby generating said wideband data signal;

a combiner, coupled to said wideband data generator, for combining said wideband data signal with said digital FM audio signal, thereby producing a composite digital FM signal; and

a mode switch, coupled to said combiner, for receiving said composite digital FM signal and a CDMA data signal and providing said composite digital FM signal to a transmitter when in said FM mode and providing said CDMA data signal to said transmitter when in said CDMA mode, said transmitter for upconverting and transmitting said information signal.

2. The dual-mode transmitter of claim 1 further comprising:

an audio interpolation filter, coupled to said digital signal processor and said combiner and interposed therebetween, for upsampling said digital FM audio signal; and

a multiplier, coupled to said, combiner and said mode switch and interposed therebetween, for scaling said composite digital FM signal.

3. A dual-mode receiver for receiving an information signal, said information signal representing a composite digital FM audio and wideband data signal when in an FM mode, and said information signal representing a code division multiple access (CDMA) data signal when in a CDMA mode, comprising:

a downconverter for downconverting said information signal, said downconverter producing in-phase (I) and quadrature-phase (Q) analog FM signals when in said FM mode, said I and Q analog FM signals being offset from zero frequency by a predetermined offset margin, and producing in-phase (I) and quadrature-phase (Q) analog CDMA signals when in said CDMA mode; and

a dual-mode interface, coupled to said downconverter, for converting said I and Q analog FM signals to I and Q digital FM signals and for converting said I and Q analog CDMA signals to I and Q digital CDMA signals, said dual-mode interface further for routing said I and Q digital FM signals to a digital FM demodulator, and for routing said I and Q digital CDMA signals to a CDMA demodulator;

said CDMA demodulator, being coupled to said dual-mode interface, for digitally demodulating said I and Q digital CDMA signals, thereby recovering said CDMA data signal;

said digital FM demodulator, being coupled to said dual-mode interface, for digitally demodulating said I and Q digital FM signals, thereby recovering said composite digital FM audio and wideband data sin said digital FM demodulator comprising

I and Q DC offset loop filters for digitally filtering said I and Q digital FM signals, eliminating said predetermined offset margin and producing I and Q digital FM baseband signals, each of said I and Q DC offset loop filters comprising

a first extraction register for extracting a most-significant bit of said I or Q digital FM signal;

a shift register, coupled to said first extraction register, for adjusting a time constant of said DC offset loop filter;

an accumulator, coupled to said shift resister, for accumulating said extracted most-significant bits;

a second extraction register, coupled to said accumulator, for extracting a predetermined number of significant bits from said accumulated most-significant bits; and

a converter, coupled to said second extraction register, for converting said extracted predetermined number of significant bits to an analog DC offset correction signal;

a digital automatic gain control network for measuring a received signal strength of said I and Q digital FM baseband signals and for altering a variable gain of said receiver in response to said received signal strength measurement; and

a frequency tracking loop filter for digitally filtering said composite digital FM audio signal and wideband data signal and for adjusting a local oscillator frequency of said receiver in response to said digitally filtered composite digital FM audio signal and wideband data signal;

wideband data recovery network, coupled to said digital FM demodulator, for recovering said wideband data signal from said composite digital FM audio and wideband data signal; and

a wideband message decoder, coupled to said wideband data recovery network, for decoding said wideband data signal.

4. The dual-mode receiver of claim 3 wherein said digital automatic gain control network comprises:

a received signal strength measurement circuit for calculating a received signal strength of said I and Q digital FM baseband signals based at least on a ratio of a maximum of an absolute value of either said I digital FM baseband signal or said Q digital FM baseband signal to a minimum of said absolute value of either said I digital FM baseband signal or said Q digital FM baseband signal;

a digital subtractor, coupled to said received signal strength measurement circuit, for subtracting a reference level from said calculated received signal strength, said digital subtractor producing a digital AGC error signal;

an integrator, coupled to said digital subtractor, for integrating said digital AGC error signal between an upper saturation limit and a lower saturation limit; and

a converter, coupled to said integrator, for converting the integrated digital AGC error signal to an analog AGC error signal.

5. The dual-mode receiver of claim 3 wherein said frequency tracking loop filter comprises:

a digital subtractor for subtracting a reference frequency from a frequency of said

composite digital FM audio signal and wideband data signal, thereby producing a digital frequency error signal;

a shift register, coupled to said digital subtractor, for adjusting a time constant of said frequency tracking loop filter in response to a gain signal;

an accumulator, coupled to said shift register, for accumulating the shifted digital frequency error signal; and

a converter, coupled to said accumulator, for converting the accumulated shifted, digital frequency error signal to an analog frequency error signal.

6. The dual-mode receiver of claim 3 wherein said wideband data recovery network comprises:

a digital phase-lock loop for recovering a Manchester-encoded non-return to zero (NRZ) data signal from a composite digital FM audio and wideband data signal; and

a NRZ decoder for decoding said Manchester-encoded NRZ data signal.

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L11: Entry 6 of 60 File: USPT Apr 15, 2003

DOCUMENT-IDENTIFIER: US 6549151 B1
TITLE: Method and apparatus for acquiring wide-band pseudorandom noise encoded waveforms

Brief Summary Text (15):
In one configuration, the analysis filters and synthesis filters are represented in a special form known as the Polyphase representation. In this form, Noble identities are can be used to losslessly move the decimators to the left of the analysis filters and the interpolators to the right of the synthesis filters.

Detailed Description Text (4):
The filters can be analog or digital depending on the type of signal 40 or the processed signal segments 56a-n. Examples of suitable analog analysis and synthesis filters include a suitably configured bandpass filter formed by one or more low pass filters, one or more high pass filters, a combination of band reject and low pass filters, a combination of band reject and high pass filters, or one or more band reject filters. Digital analysis and synthesis filters are typically defined by software architecture that provides the desired filter response.

Detailed Description Text (16):
Noble identities can be used to losslessly move the decimators to the left of the analysis filters and the L-fold up-sampler and/or expander to the right of the synthesis filters. In this manner, the analysis and synthesis filters operate on lower rate data, thereby realizing significant computational savings. The noble identities include:

Detailed Description Text (20):
Based on the foregoing, FIG. 6 is a polyphase representation based implementation of H(z) without noble identities and FIG. 7 is a polyphase representation-based implementation of the analysis filters H(z) using noble identities to move the decimators ahead of the analysis filters. In this configuration, H.sub.o (z.sup.2) and H.sub.1 (z.sup.2) are FIR filters of order n.sub.o +1 and n.sub.1 +1, where N=n.sub.0 +n.sub.1 +1. H.sub.o (z.sup.2) and H.sub.1 (z.sup.2) operate at half the rate as compared to H(z) and therefore have two units of time in which to perform all the necessary computations, and the components are continually active (i.e., there are no resting times). Accordingly, there is an M-fold reduction in the number of multiplications and additions per unit of time when using both polyphase representation and the noble identities to implement an M-fold decimation filter.

Detailed Description Text (21):
Subband signal processing can take a variety of forms. In one embodiment shown in FIG. 8 which depicts a receiver and antenna architecture, the source signal 40 and subband signals 48a-n are in analog form and a plurality of quantizers or analog-to-digital converters are used to convert the subband signals 48a-n to digital form before further processing 82 (e.g., correlation for encoded subband signals, subband signal digital beamforming in multiple antenna systems, etc.) and/or synthesis of the digital subband signals 78a-n is performed. As noted above, the subband signals 48a-n are preferably sampled by each of the decimators or downsamplers 64a-n at a rate of at least about twice the bandwidth of the corresponding subband signal 48a-n to maintain fidelity. As shown in FIG. 9, each quantizer, or analog-to-digital converter, 74a-n determines the digital word or representation 90a-n that corresponds to the bin 86a-n having boundaries capturing the amplitude of the analog subband signal at that time and outputs the digital word or representation that represents the selected quantization level assigned to the respective bin. The digital subband signals 78a-n are converted 94a-n from radio

frequency (RF) to base band frequency and optionally subjected to further signal processing 60. The processed subband signals 98 are formed into a digital composite signal 102 by the synthesis filter bank 60.

Detailed Description Text (57):
In one variation of the system of FIG. 15 that is depicted in FIGS. 17-18, multiplexed radar transmitted receiver architectures are depicted. The radar signals 400a-n are a number of coded waveforms that operate in separate, contiguous subbands (referred to as "radar subband signals"). As shown in FIG. 17, the radar signals 40 are simultaneously transmitted by a plurality of transmitters 404a-n that each include a plurality of analysis filters (not shown) to form the various radar subband signals 400a-n. Referring to FIG. 18, the various radar subband signals 400a-n are received by a signal receptor 410 and passed through a plurality of bandpass filters 414a-n. A bandpass filter 414a-n having unique bandpass characteristics corresponds to each of the radar subband signals. The various filtered subband signals 416a-n are sampled by a plurality of decimators 422a-n and quantized by a plurality of quantizers 426a-n to form digital subband signals 430a-n. The digital subband signals 430a-n are analyzed by a plurality of detectors 434a-n to form a corresponding plurality of detected signals 438a-n. The detectors 434a-n use a differently coded waveform for each of the transmitted radar subband signals 400a-n so that the subband radar signals can be individually separated upon reception. As noted above in FIGS. 14-15, the coded radar waveform is decomposed by a plurality of analysis filters (not shown) that are identical to the analysis filters in the receiver to provide replicated subband signals to the detectors 434a-n. Each detector 434a-n correlates a radar subband signal 430a-n with its corresponding replicated subband signal to form a plurality of corresponding detected signals 438a-n. The detected signals 438a-n are analyzed by a synthesis filter bank 412a-n to form a composite radar signal 446.

L11: Entry 16 of 60 File: USPT Jun 26, 2001

DOCUMENT-IDENTIFIER: US 6252535 B1
**** See image for Certificate of Correction ****
TITLE: Method and apparatus for acquiring wide-band pseudorandom noise encoded waveforms

Brief Summary Text (18):
In one configuration, the analysis filters and synthesis filters are represented in a special form known as the Polyphase representation. In this form, Noble identities are can be used to losslessly move the decimators to the left of the analysis filters and the interpolators to the right of the synthesis filters.

Detailed Description Text (4):
The filters can be analog or digital depending on the type of signal 40 or the processed signal segments 56a-n. Examples of suitable analog analysis and synthesis filters include a suitably configured bandpass filter formed by one or more low pass filters, one or more high pass filters, a combination of band reject and low pass filters, a combination of band reject and high pass filters, or one or more band reject filters. Digital analysis and synthesis filters are typically defined by software architecture that provides the desired filter response.

Detailed Description Text (20):
Noble identities can be used to losslessly move the decimators to the left of the analysis filters and the L-fold up-sampler and/or expander to the right of the synthesis filters. In this manner, the analysis and synthesis filters operate on lower rate data, thereby realizing significant computational savings. The noble identities include:

Detailed Description Text (24):
Based on the foregoing, FIG. 6 is a polyphase representation based implementation of H(z) without noble identities and FIG. 7 is a polyphase representation-based implementation of the analysis filters H(z) using noble identities to move the decimators ahead of the analysis filters. In this configuration, H.sub.o (z.sup.2) and H.sub.1 (z.sup.2) are FIR filters of order n.sub.o +1 and n.sub.1 +1, where N=n.sub.o +n.sub.1 +1. H.sub.o (z.sup.2) and H.sub.1 (z.sup.2) operate at half the rate as compared to H(z) and therefore have two units of time in which to perform all the necessary computations, and the components are continually active (i.e., there are no resting times). Accordingly, there is an M-fold reduction in the number of multiplications and additions per unit of time when using both polyphase representation and the noble identities to implement an M-fold decimation filter.

Detailed Description Text (25):
Subband signal processing can take a variety of forms. In one embodiment shown in FIG. 8 which depicts a receiver and antenna architecture, the source signal 40 and subband signals 48a-n are in analog form and a plurality of quantizers or analog-to-digital converters are used to convert the subband signals 48a-n to digital form before further processing 82 (e.g., correlation for encoded subband signals, subband signal digital beamforming in multiple antenna systems, etc.) and/or synthesis of the digital subband signals 78a-n is performed. As noted above, the subband signals 48a-n are preferably sampled by each of the decimators or downsamplers 64a-n at a rate of at least about twice the bandwidth of the corresponding subband signal 48a-n to maintain fidelity. As shown in FIG. 9, each quantizer, or analog-to-digital converter, 74a-n determines the digital word or representation 90a-n that corresponds to the bin 86a-n having boundaries capturing the amplitude of the analog subband signal at that time and outputs the digital word or representation that represents the selected quantization level assigned to the

respective bin. The digital subband signals 78a-n are converted 94a-n from radio frequency (RF) to base band frequency and optionally subjected to further signal processing 60. The processed subband signals 98 are formed into a digital composite signal 102 by the synthesis filter bank 60.

Detailed Description Text (72):
In one variation of the system of FIG. 15 that is depicted in FIGS. 17-18, multiplexed radar transmitter and receiver architectures are depicted. The radar signals 400a-n are a number of coded waveforms that operate in separate, contiguous subbands (referred to as "radar subband signals"). As shown in FIG. 17, the radar signals 400a-n are simultaneously transmitted by a plurality of transmitters 404a-n that each include a plurality of analysis filters (not shown) to form the various radar subband signals 400a-n. Referring to FIG. 18, the various radar subband signals 400a-n are received by a signal receptor 410 and passed through a plurality of bandpass filters 414a-n. A bandpass filter 414a-n having unique bandpass characteristics corresponds to each of the radar subband signals. The various filtered subband signals 416a-n are sampled by a plurality of decimators 422a-n and quantized by a plurality of quantizers 426a-n to form digital subband signals 430a-n. The digital subband signals 430a-n are analyzed by a plurality of detectors 434a-n to form a corresponding plurality of detected signals 438a-n. The detectors 434a-n use a differently coded waveform for each of the transmitted radar subband signals 400a-n so that the subband radar signals can be individually separated upon reception. As noted above in FIGS. 14-15, the coded radar waveform is decomposed by a plurality of analysis filters (not shown) that are identical to the analysis filters in the receiver to provide replicated subband signals to the detectors 434a-n. Each detector 434a-n correlates a radar subband signal 430a-n with its corresponding replicated subband signal to form a plurality of corresponding detected signals 438a-n. The detected signals 438a-n are analyzed by a synthesis filter bank 412a-n to form a composite radar signal 446.

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L2: Entry 29 of 35 File: USPT Apr 1, 1997

DOCUMENT-IDENTIFIER: US 5617060 A
TITLE: Method and apparatus for automatic gain control and DC offset cancellation in quadrature receiver

Abstract Text (1):
An automatic gain control (AGC) and D.C. offset correction method and apparatus for controlling signal power of a received RF signal within a dual mode quadrature receiver is disclosed herein. In a preferred implementation the automatic gain control apparatus may be adjusted to provide a desired control response to various fading characteristics of a received FM, FSK, GMSK, QPSK, or BPSK signal. The AGC apparatus includes an adjustable gain amplifier having an input port for receiving an input signal, a control port for receiving a gain control signal, and an output port for providing an output signal. A quadrature downconverter coupled to the output port serves to translate the frequency of the output signal to a baseband frequency, thereby producing baseband signals. In a preferred implementation the downconverter is operative to map the carrier frequency of the output signal to a baseband frequency offset by a predetermined margin from D.C. Two high gain active lowpass filters provide out-of-band signal rejection for the baseband signals. A D.C. feedthrough suppression loop, disposed to receive said baseband signal, suppresses D.C. offsets produced by the downconverter and lowpass filters, hence providing a compensated baseband signal. The AGC apparatus is further disposed to generate a received power signal based on the power of the output signal. A saturating integrator compares the received power signal to a reference signal and produces the gain control signal by integrating or by refraining from integration based on values of the reference, received power signal, and gain control signals, thereby extending the usable dynamic range of the receiver for FM mode.

Brief Summary Text (5):
In analog receivers, such as are used in narrowband FM cellular communication systems, FM demodulators are employed to extract information encoded in the phase of an incident waveform. Existing FM demodulators often include an analog frequency discriminator preceded by an analog limiter, with the limiter serving to constrain the input signal power to a constant level. In this way maximum signal to noise ratio is maintained at the input to the frequency discriminator over the full dynamic range of the FM input signal. However, such an analog signal processing technique generally involves extensive signal filtering, and frequently is implemented using a large number of discrete components. Moreover, it has been demonstrated that improved performance may be achieved using linear digital waveform demodulation rather than analog demodulation. Unfortunately, conventional demodulation techniques are often not applicable to digital receivers, since clipping of the received signal would result in corruption of the data derived therefrom.

Brief Summary Text (7):
In the cellular environment, a digital receiver may receive a signal which experiences rapid and wide variations in signal power. In digital receivers such as are used in a code division multiple access (CDMA) and Time Division Multiple Access (TDMA) mobile cellular telephone, it is necessary to control the power of the demodulated signal for proper signal processing. However, in digital receivers to be both CDMA or TDMA compatible and conventional FM compatible, i.e., dual-mode digital/FM receivers, it is necessary to provide power control of both wideband CDMA (or TDMA) signals and narrowband FM signals. The control process is complicated by the differing dynamic ranges associated with the received FM and CDMA signal power. That is, the magnitude of received FM signals may vary over a dynamic range greater than 100 dB, whereas CDMA systems typically result in a more limited dynamic range,

i.e., approximately 80 dB.

Brief Summary Text (9):
It would also be desirable to provide digital AGC in inexpensive receivers utilizing analog to digital (A/D) converters with limited dynamic range. Again, because FM signals within cellular systems may vary more than 100 dB and relatively inexpensive 8-bit A/D's are limited to a dynamic range of approximately 48 dB, a cost effective AGC implementation should be capable of controlling the gain of the portion of the receiver preceding the A/D converters so as to control the signal's dynamic range at the A/D converter. The alternative is to employ expensive A/D converters having greater dynamic range, thereby increasing the cost of the receiver or to increase the AGC range of the analog portion of the radio which is very difficult and costly.

Brief Summary Text (11):
In standard FM cellular telephones, the AGC function is performed by a circuit called a limiter. When a limiter is used, out-of-band signal rejection can only be done using intermediate frequency (IF) filters. Although the requisite signal rejection capability may be achieved through the use of ceramic IF filters, these tend to be relatively large and expensive. Smaller and less expensive IF filters are generally incapable of being realized so as to possess the desired signal rejection characteristics, and hence are generally not employed in FM cellular telephone receivers.

Brief Summary Text (14):
However, for constant amplitude modulations such as FM and continuous phase FSK (which are used in FM cellular telephone systems such as AMPS) and Gaussian Minimum Shift Keying (GMSK) (used in some TDMA systems), the carrier must be preserved in order to demodulate the received signal.

Brief Summary Text (15):
The employment of active baseband IC filters leads to the necessity of providing some mechanism for suppression of undesired D.C. offsets. The IF processing chain of conventional digital cellular telephone receivers typically includes a local oscillator (L.O.) having a frequency selected such that the carrier frequency is downconverted to D.C., and a simple D.C. notch filter is used to remove unwanted D.C. offsets. If an FM, FSK, or GMSK signal is processed by such an IF processing chain, then the D.C. offset suppression will not only remove unwanted D.C. components, but also critical phase and amplitude information at the carrier frequency. That is, in FM cellular telephone systems significant amplitude and phase information is present at the carrier frequency, and performance will be adversely affected if such information is destroyed.

Brief Summary Text (16):
However, there are two narrow bands of frequencies in between the carrier frequency F.sub.c and F.sub.c +F.sub.1 and between F.sub.c and F.sub.c -F.sub.1 (where F.sub.1 is the lowest frequency expected in the demodulated spectrum, typically F.sub.1 =300 Hz for FM cellular) which can be suppressed without adversely affecting the demodulated signal. Although minimal voice information is carried at intermodulation products at frequencies close to the carrier frequency, such products are uncommon and of relatively short duration. Accordingly, the suppression of only the low-frequency intermodulation products after baseband downconversion does not usually result in the loss of appreciable voice information. Similarly, in FSK and GMSK systems, very little signal power is present below F.sub.1 =(symbol rate)/100, so again the frequency band between F.sub.c and F.sub.c +F.sub.1 may be suppressed without degradation of the digital data.

Brief Summary Text (19):
The present invention is a novel automatic gain control method and apparatus for controlling signal power of a received RF signal over a wide dynamic range. In a preferred implementation the automatic gain control apparatus may be adjusted to provide a desired control response to various fading characteristics of the received RF signal. In applications where the signal of interest is a suppressed carrier digital format such as BPSK or QPSK (for CDMA Digital Cellular) or a constant envelope continuous-phase format such as GMSK, FSK, or FM (used in AMPS cellular

phase system), the apparatus of the present invention is capable of providing the necessary gain control, out-of-band signal rejection, and downconversion to baseband, with no D.C. offset.

Brief Summary Text (20):
In accordance with the present invention an automatic gain control (AGC) apparatus for a dual mode receiver is disclosed. The AGC apparatus includes an adjustable gain amplifier having an input port for receiving an input signal, a control port for receiving a gain control signal, and an output port for providing an output signal. A downconverter coupled to the output port serves to translate the frequency of the output signal to a baseband frequency, thereby producing a baseband signal. In a preferred implementation the downconverter is operative to map the carrier frequency of the received signal of the output signal to a baseband frequency offset by a predetermined margin from D.C. A D.C. feedthrough suppression loop, disposed to receive said baseband signal, suppresses D.C. feedthrough signals produced by the downconverter, hence providing a compensated baseband signal.

Detailed Description Text (2):
In a digital receiver, such as used in a code division multiple access (CDMA) portable cellular communications device, it is necessary to set the power of the processed signal to a constant level. In the cellular environment, a receiver may receive a signal which experiences rapid and wide variations in signal power. In order to properly process the digital data contained within the received signal the signal power must be controlled within the receiver. In a dual-mode digital receiver, e.g., a digital receiver capable of processing both CDMA (or TDMA) and standard FM signals, the received signal dynamic range will vary as a function of the selected operative mode. Accordingly, an automatic gain control apparatus for a digital receiver is disclosed which is capable, in each of its operative modes, of compensating for variation in received signal power in either environment.

Detailed Description Text (3):
FIG. 1 illustrates in block diagram form an exemplary application of the automatic gain control apparatus of the present invention. In FIG. 1, the automatic gain control apparatus is implemented in the transceiver of a CDMA portable cellular telephone 10. Telephone 10 may be a dual mode, i.e. CDMA (or TDMA) and conventional FM compatible. The automatic gain control apparatus of the present invention is capable of providing power control of both wideband CDMA (or TDMA) signals and narrowband FM signals. The compatibility of such circuitry to operate on both wideband and narrowband signals provides cost, component and power savings for the receiver.

Detailed Description Text (7):
The gain controlled IF signals are output from amplifier 18 to a second frequency downconverter, downconverter 20, where the IF signals are converted to a lower frequency range and are provided as corresponding in-phase and quadrature-phase baseband signals I.sub.BB and Q.sub.BB. In the embodiment shown in FIG. 1, the baseband signals in the CDMA mode of operation are I and Q samples of encoded digital data which are output for further phase demodulation and correlation. In a dual mode receiver, downconverter 20 also frequency downconverts FM signals so as to provide baseband FM in-phase and quadrature-phase signals, which are further phase/frequency demodulated into an audio output signal.

Detailed Description Text (15):
Again referring to FIG. 3, an embodiment of a switching arrangement is shown using RF switches 49 and 55. RF switches 49 and 55 couple CDMA IF bandpass filter 51 to IF amplifier 18 during CDMA mode as shown by the setting of the switches in FIG. 3. In FM mode, the position of RF switches 49 and 55 changes to couple FM IF bandpass filter 53 and limiter 54 to IF amplifier 18. FM IF bandpass filter 53 for rejecting out-of-channel interference defines the bandwidth of the FM signals provided through limiter 54 to amplifier 18. For example, in FM mode operation the FM IF filter 53 is designed to have a passband spanning approximately one cellular channel (e.g., 30 kHz), and a stopband extending significantly beyond (e.g., .+-.60 kHz) the IF center frequency. During CDMA mode operation the CDMA IF filter 51 is designed to reject out-of-channel interference and defines the bandwidth of the CDMA signals provided to amplifier 18. For example during CDMA mode, CDMA IF bandpass filter 51 may

provide a passband commensurate with the chip rate of the baseband portion of the receiver (e.g. 1.26 MHz), and provide a predefined rejection bandwidth (e.g. 1.8 MHz). In an alternative embodiment, limiter 54 could be in the common path before IF amplifier 18.

CLAIMS:

13. The automatic gain control apparatus of claim 12 wherein the apparatus operates in either a code division multiple access mode (CDMA) or a frequency modulated (FM) mode and the at least one low pass filter is comprised of a first filter for operation in the CDMA mode and a second filter for operation in the FM mode.